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Comments on MM Docket No. 93-225
Means of Measuring Modulation in Broadcast

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Introduction

The Commission has asked for comments regarding potential "improvements" in instruments for measuring modulation of FM broadcast stations. This writer has had many years experience designing audio processors for broadcast, and has more recently researched the transient response of FM exciters, composite STLs, and modulation monitors. This research revealed that many commercial exciters and STLs do not accurately reproduce the waveforms applied to their inputs. Further, many commercial modulation monitors do not accurately indicate the peak modulation of the FM signal. In all cases, these problems proved to be caused by inadequate low-frequency response, and not by high-frequency ringing and overshoot, as many had previously believed.[1]

Effect of Subcarriers on "Peak-Weighted" Modulation Measurement

This writer is a consultant to AKG Acoustics, and fully concurs with the Comments filed by Robert Orban, its Chief Engineer. To complement Orban's comments, this writer would like to present the results of certain measurements indicating that "peak-weighted" measurements being made today can significantly under-indicate total modulation in the presence of subcarriers, permitting modulation levels contrary to the Commission's desires when it changed the modulation rules to permit 100% plus one-half the total subcarrier modulation, provided that the total peak modulation never exceeds 110%.[2]

Table 1 shows the results of this writer's measurements. In all cases, the stereo generator was modulated with normal stereophonic program material as processed by an Orban 8100A+XT2 audio processing system with the optional "Card #0," which provides additional spectral protection above 60kHz in the baseband. The audio processor was set to its factory-recommended settings for "moderate" processing. Pilot injection was 9%, and the modulation level was adjusted to produce a total peak deviation of ±75kHz, which was considered to be 100%. A DC-coupled oscilloscope connected to the baseband output of the processing system showed peaks to be tightly constrained to the desired modulation, with less than 2% overshoot.

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[1] Greg J. Ogonowski, "A New Approach To FM Composite Baseband Overshoot Control," *Proc. 1990 NAB Engineering Conference,* Atlanta, GA, April 1990.

[2] Michael C. Rau, Harrison J. Klein, and John Kean, "Increased FM Deviation, Additional Subcarriers and FM Broadcasting: A Technical Report.", NAB, Westinghouse Broadcasting & Cable, Inc., and NPR, August 30, 1983.

The output of the processor was applied to a wideband DC-coupled summing amplifier, which allowed the subcarriers to be added to the output of the audio processor. The output of the summing amplifier was applied to the input of a Belar "Wizard" modulation analyzer, eliminating any possibility of errors caused by RF modulation and demodulation.

The Belar "Wizard" can be adjusted to respond to instantaneous peak modulation, and can also be set to provide a time-weighted peak indication. For these tests, the time-weighting was set to 900µs, corresponding to 9 cycles of a 10kHz tone burst, as implied in the old FCC Rules regarding modulation measuring instruments.

In *Table 1,* one can see that the "weighted peak" measurement was anywhere from 3% to 6% low by comparison to the instantaneous peak measurement. Even the instantaneous peak measurement did not necessarily agree with a simple arithmetic addition of the various amplitudes of the various elements in the baseband. For example, in a somewhat artificial situation, the 57kHz and 95kHz were phase-locked to the pilot so that the zero-crossings of the 19kHz, 57kHz, and 95kHz waves coincided. Because the 57kHz and 95kHz waves are, respectively, the third and fifth harmonics of the pilot, this created a somewhat square wave-like signal with a true peak level 7% less than one would expect by simple arithmetic addition of the various amplitudes. Instead of 120% (100%+10%+10%), the unweighted peak level was, accurately, 113%, and the weighted peak level was 110%.

In "real-world" cases, where the subcarriers were not phase-locked to the pilot, the unweighted peak measurement agreed exactly with arithmetic addition. In these cases, the weighted peak measurement was as much as 6% low, which would cause the station using such measurements to modulate substantially beyond the limits established by the Commission on the basis of the Rau, Klein, and Kean research. These results were obtained *regardless* of whether "light" or "heavy" audio processing was used. Since this abuse is possible while staying within the weighted peak limits implied by the old FCC Rules, one could imagine even worse abuses should the requirements be further relaxed.

Conclusions

Significantly, with a perfect simulated transmission path, there was *no difference* between the weighted and unweighted measurements of program modulation in absence of subcarriers. This indicates that, *if the transmission system has accurate transient response,* extending the peak weighting of the indicator to 900µs provides no increase in potential modulation in the absence of subcarriers, regardless of whether "light" or "heavy" audio processing is used. The only effect of 900µs-weighted metering in a properly-operating transmission system is its ability to under-read modulation caused by subcarriers, which thwarts the Commission's intent in implementing Rau, Klein, and Kean's conclusions.

[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]	[illegible]
0%	0%	0%	0%	100%	100%	OFF
0%	0%	0%	0%	100%	100%	ON
10%	0%	0%	10%	100%	113%	OFF
10%	0%	0%	10%	100%	110%	ON
0%	10%	10%	0%	100%	120%	OFF
0%	10%	10%	0%	100%	114%	ON
0%	10%	0%	0%	100%	110%	OFF
0%	10%	0%	0%	100%	106%	ON
0%	0%	10%	0%	100%	110%	OFF
0%	0%	10%	0%	100%	105%	ON
0%	20%	0%	0%	100%	120%	OFF
0%	20%	0%	0%	100%	116%	ON
0%	0%	20%	0%	100%	120%	OFF
0%	0%	20%	0%	100%	115%	ON

Table 1
Effect of peak-weighted modulation measurement on composite baseband with subcarriers

A NEW APPROACH TO FM COMPOSITE BASEBAND OVERSHOOT CONTROL

ACHIEVING MAXIMUM MODULATION AND EFFECTIVE LOUDNESS IN FM STEREO BROADCASTING

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PORTIONS REPRINTED FROM **1990 NAB ENGINEERING CONFERENCE PROCEEDINGS**

INTRODUCTION

It is well known that processed, peak controlled program material causes most composite STL systems and FM exciters to produce overshoot in the FM composite baseband signal. This overshoot occurs even in properly band-limited systems that utilize STL paths without multipath. Accordingly, loudness is compromised because average modulation must be reduced to prevent illegal peak overmodulation caused by this overshoot.

Previous attempts to eliminate this overshoot have degraded system performance. We have developed a new approach that minimizes this overshoot without compromise.

THE FM STEREO SYSTEM

The world-standard FM stereo "pilot-tone" system encodes the sum of the channels (L+R) in the frequency range of 30-15,000Hz in the stereo baseband – the "stereo main channel." It encodes the difference between the channels (L-R) on a double-sideband suppressed-carrier sub-channel centered at 38kHz, and occupying 23kHz to 53kHz in the stereo baseband – the "stereo sub-channel." A pilot tone at 19kHz tells the receiver that a stereophonic transmission is being received, and provides a phase and frequency reference to permit the receiver to regenerate the 38kHz sub-carrier to use in its stereo demodulator.

Any energy that appears in the frequency range from 30 to 19,000Hz caused by a signal in the stereo sub-channel is termed "sub-channel-to-main channel crosstalk." Any energy that appears in the frequency range from 19 to 57kHz caused by a signal in the stereo main channel is termed "main-channel-to-sub-channel crosstalk."

When the stereo encoder is driven by a pure right-only or left-only signal, "stereophonic separation" can be measured at the stereo decoder as the ratio between the desired and undesired signal levels, where the "desired" signal is the signal appearing in the decoder output channel corresponding to the channel driven at the encoder, and the "undesired" signal is the signal caused by the desired signal that appears at the remaining output.

Ideally, crosstalk is non-existent and stereophonic separation is infinite. In practice, both linear and non-linear errors cause these characteristics to deteriorate.

In the linear domain, separation and crosstalk are mathematically orthogonal. Phase and frequency errors that cause one to deteriorate will not affect the other. For example, phase or frequency errors in the composite signal channel will cause separation to deteriorate, but cannot affect crosstalk, since the stereo main and sub-channels are already separated in frequency and changes in phase or amplitude response in the composite channel cannot affect this frequency separation. Conversely, mismatches between the linear response of the left and right signal paths prior to the stereo encoder will cause crosstalk, but cannot affect separation.

Non-linearities in the composite channel can cause both separation and crosstalk to deteriorate because such errors cause harmonic and intermodulation distortion that introduce new frequencies into the baseband. These new frequencies are likely to inject power into a part of the baseband spectrum that will be decoded by the stereo decoder in spatial locations different than the locations of the original sound sources. Further, these new frequencies are perceived by the ear not as changes in spatial localization, but as highly offensive distortion.

This is somewhat analogous to "aliasing distortion" in a sample-data audio system. In such a system, any input frequencies greater than one-half of the sampling frequency (the "Nyquist frequency") are encoded with the wrong frequency: they "fold around" the Nyquist frequency and appear at the decoder as frequencies unrelated to the program material that produced them. The ear perceives this "aliasing" as offensive distortion.

PREVIOUS NON-LINEAR SOLUTION

The most common technique for reducing FM composite baseband signal overshoot has been composite baseband clipping. Composite clipping has been very controversial because it causes signal degradation and because early implementations that clipped the pilot could violate the FCC rules. No implementation prevents dynamic signal degradation.

The composite baseband clipper is a non-linear device. Thus, it generates distortion and aliasing products that contaminate the composite baseband signal, degrading dynamic stereo separation and causing dynamic audible distortion. It also produces distortion products in the subcarrier region, reducing or destroying the sub-carrier's market value and reducing revenue potential.

While it is possible to lowpass-filter the clipped baseband (thus protecting the sub-carriers), such filtering does nothing to eliminate intermodulation distortion in the stereo baseband region below 57kHz, and will also tend to increase peak modulation, partially negating the peak control provided by the composite clipper.

If the FM exciter is the source of overshoot (as opposed to the STL), or if the clipper precedes the STL, then the composite baseband clipper cannot control overshoot. Instead, it can actually *increase* overshoot because the clipping process produces increased amounts of infrasonic intermodulation distortion products.

Some have argued that composite baseband clipping increases loudness more than audio clipping in the left and right channels. But this loudness increase is accompanied by degraded dynamic stereo separation and crosstalk. To preserve dynamic stereo performance, the spectra of the stereo main channel and sub-channel must be completely isolated: the main channel must not have any energy above 19kHz, and the sub-channel must not have any energy below 19kHz.

One consequence of such frequency separation is this: in a system that achieves high dynamic separation and low crostalk, it must be impossible for the system's final filter/limiter to reproduce any approximation to a square wave if the square wave's fundamental frequency is higher than one-third the cut-off frequency of the low-pass filter prior to stereo encoding (typically 15kHz). This is because the first harmonic of the square wave is three times the frequency of the fundamental, so the low-pass filter removes it (and all higher harmonics too); any square wave above 5kHz will emerge from the receiver as a sine wave. Because they generate spurious harmonic and intermodulation products, composite baseband clippers do not meet this criterion and thus compromise dynamic stereo performance.

Composite clipping has one potential advantage. Conventional wisdom holds that the peak modulation of the composite baseband is the greater of the left or right channel levels, plus the pilot. However, this is only an approximation because the pilot is correlated in phase with the 38kHz suppressed sub-carrier. This causes the total composite modulation to decrease slightly when the left and right channels are unequal in level. Assuming 10% pilot injection and holding the left channel at 100% modulation, decreasing the right channel from 100% to 0% modulation will cause the composite modulation to decrease by 2.8%. Perfectly accurate peak limiting in the audio domain, prior to stereo encoding, can only control the composite modulation to an accuracy of -2.8%/+0%. Only a process that is aware of the total peak composite modulation (including the pilot and any subcarriers) can control composite modulation accurately. Since composite baseband clipping controls the peak deviation of the composite signal precisely (assuming the pilot is also clipped), it can theoretically be louder than peak limiting in the audio domain. But the "advantage" is an imperceptible 0.24dB at best! Composite baseband clippers that do not clip the pilot (which are the only clippers legal for use in the U.S.) do not eliminate the interleave error, and therefore produce no loudness advantage whatever!

In these digital times, bad audio quality has become unacceptable to formerly unsophisticated consumers. It is absurd that composite baseband clippers are being used to degrade system performance below that of some of the least expensive receivers! Composite clipping is a very easy, unsophisticated method of increasing apparent loudness of the broadcast signal, but it compromises quality in a way that is unacceptable to any broadcaster trying to compete with CD or the newer recordable digital media.

Left channel modulation 100%.
Right channel modulation from 0% to 100%.
10% pilot injection.
Peak deviation of composite shown with 100% normalized to L=R with pilot present.

Right % Modulation	Composite % Modulation
100.0%	100.00%
99.5%	99.79%
99.0%	99.59%
98.0%	99.26%
97.0%	98.99%
96.0%	98.77%
95.0%	98.59%
94.0%	98.45%
93.0%	98.33%
92.0%	98.22%
91.0%	98.14%
90.0%	98.06%
80.0%	97.65%
70.0%	97.48%
60.0%	97.39%
50.0%	97.33%
40.0%	97.29%
30.0%	97.26%
20.0%	97.24%
10.0%	97.22%
0.0%	97.20%

Fig. 1 INTERLEAVING ERROR IN THE FM STEREO SYSTEM

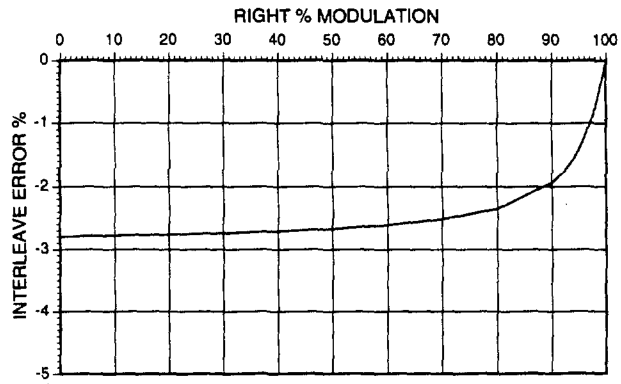

Fig. 2 INTERLEAVING ERROR IN THE FM STEREO SYSTEM

Fig. 1 indicates the exact interleaving failure due to pilot summation in the FM stereo system.* Fig. 2 is a plot of the interleaving error in % of modulation versus right channel modulation., with the left channel modulation held constant at 100%.

* Robert Orban, private communication.

OVERSHOOT SOURCE

Extensive computer modeling and analysis of several current-generation composite STL systems and FM exciters has revealed that the overshoot problem is not in the high frequency domain (as previously assumed), but instead at infrasonic frequencies. *All* (except one) of the systems modeled have infrasonic peaks in their frequency response, and/or have insufficient low frequency response to accurately reproduce a processed composite baseband signal. Some of the systems even suffer from marked non-linearity, having different frequency responce at different modulation levels at very low frequencies, aggravating the problem. This poor low frequency transient response can be caused by incorrectly designed AFC loops and/or deficient low frequency response of the composite baseband amplifiers. Figs. 3 and 4 show the system response of two popular composite STL systems. Figs. 5 and 6 show the system response of two popular FM exciters. Note the radical differences in low frequency response, and how the response of a given system depends on its RF operating frequency. It's no surprise or myth that each of these modulators has its own sonic signature as well. (This might explain why legend has it that certain FM channels sound better than others!)

Another important consideration is the overall system performance of the various components in the composite signal path. When these components are cascaded, the system response deteriorates. The amount of degradation depends on input and output impedance effects of the composite baseband amplifiers and how many systems are cascaded, so the audible performance of a system is not always simply the sum of the performance of its parts. Fig. 7 shows the STL 1 and Exciter 1 total system response. (Once again, this might explain why legend contends, that certain combinations of equipment sound better than others!)

PEAK-WEIGHTED MONITORING

Recently, on the basis of a very controversial interpretation of the FCC Rules & Regulations, devices have been introduced to change the way that broadcasters measure modulation. By delaying the response of the peak indicator, the modulation monitor ignores peaks of less than 1 millisecond duration (assumed to be overshoots), and does not indicate over-modulation under these conditions. Although this technique of modulation measurement is under close investigation by the industry (for reasons not relevant here), it does not ignore the type of composite baseband overshoot described above. Instead, provided the peak indicator circuitry has been designed correctly, the modulation monitor accurately measures this overshoot because its duration is far longer than the delay of the peak indicator. So the composite path must still be free from infrasonic overshoot to preserve peak control providing maximum loudness with minimum distortion.

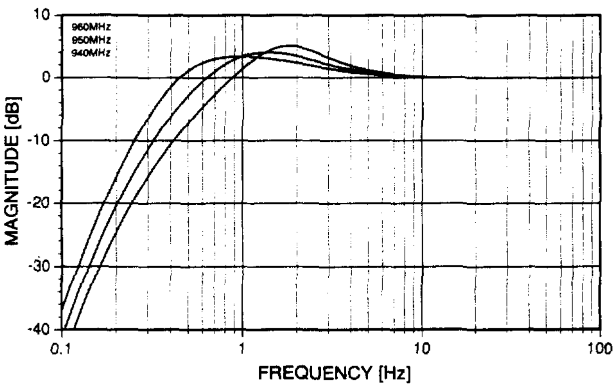

Fig. 3 STL SYSTEM 1 INFRASONIC FREQUENCY RESPONSE
MOSELEY ASSOCIATES, INC. PCL-606/C

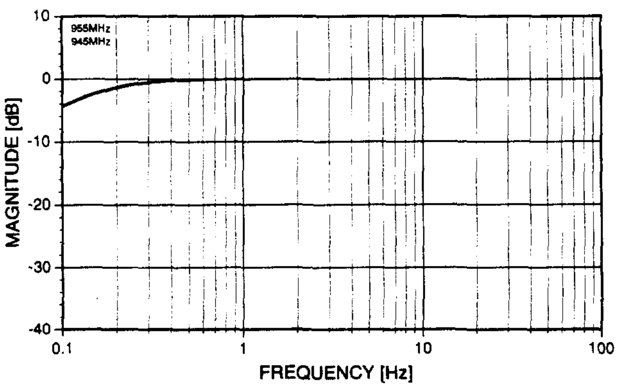

Fig. 4 STL SYSTEM 2 INFRASONIC FREQUENCY RESPONSE
TFT, INC. 8300

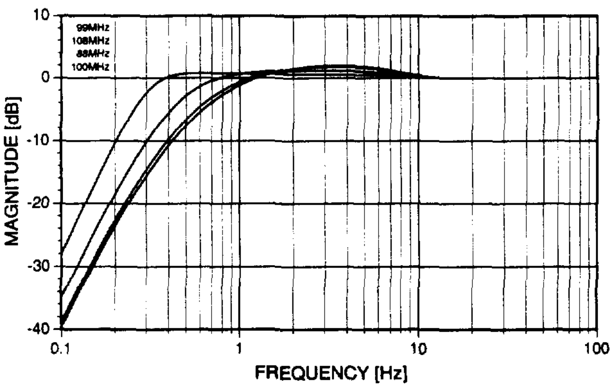

Fig. 5 FM EXCITER 1 INFRASONIC FREQUENCY RESPONSE
CONTINENTAL ELECTRONICS MFG., CO., INC. 802A

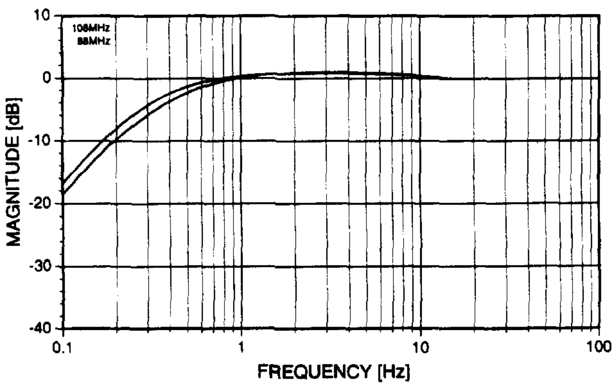

Fig. 6 FM EXCITER 2 INFRASONIC FREQUENCY RESPONSE
BROADCAST ELECTRONICS, INC. FX-50

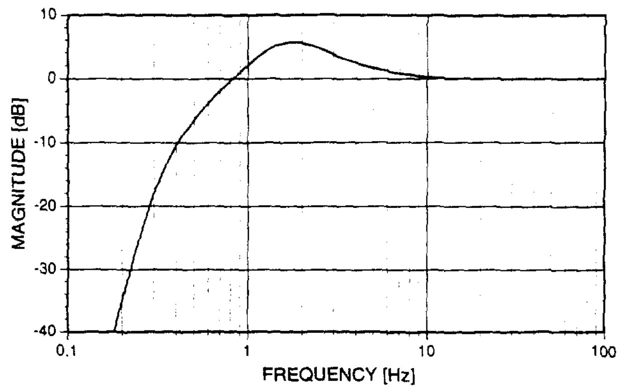

Fig. 7 STL SYSTEM 1 AND EXCITER 1 TOTAL SYSTEM RESPONSE

RESPONSE REQUIREMENTS

If we model the system (to a first-order approximation), as a high-pass filter with a single dominant pole, we can use the equation in Fig. 8 to compute the percentage of overshoot when a square-wave of frequency F (Hz) is applied to the system input. This overshoot will occur regardless of whether the low-frequency rolloff is in the composite channel, or in the left and right audio channels after peak limiting. In the former case, the rolloff can also compromise low-frequency separation. To achieve less than 1% overshoot with a 50Hz square-wave (a reasonable criterion for good peak control), the dominant pole must be located at 0.16Hz or lower with no peaking! Good 10Hz square-wave response does not predict low overshoot because a peak in the region below 10Hz can phase equalize the 10Hz fundamental, while simultaneously distorting the phase and amplitude of the components below 10Hz. If more than one low-frequency roll-off element is cascaded in the composite path, each element's cut-off frequency must be substantially below 0.16Hz. Fig. 9 shows the minimum required low-frequency response.

$$\text{Overshoot \%} = 100\,[1 - \mathit{exp}(P/2F)\,]$$

where $P = -(1/RC)$
where F = Frequency in Hz

Fig.8 OVERSHOOT EXPRESSION

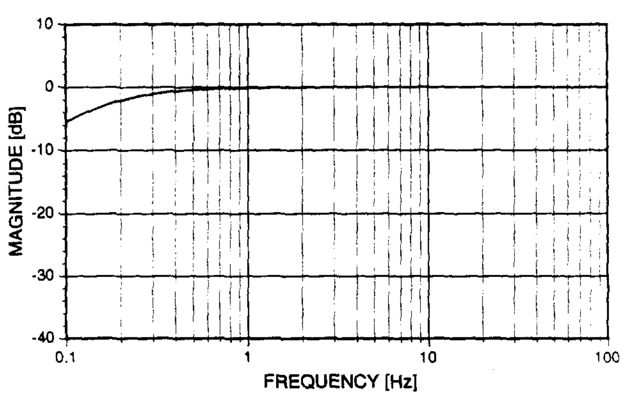

Fig. 9 MINIMUM LOW-FREQUENCY RESPONSE REQUIREMENT

NEW LINEAR SOLUTION

Since poor infrasonic frequency response causes composite baseband overshoot, it can be eliminated by the proper design of the AFC circuitry and composite baseband amplifiers. It can also be corrected, with a specially designed infrasonic equalizer, although not as accurate. Both methods are linear solutions to linear problems, with the first method preferred. Unlike composite baseband clippers, linear correction produces no distortion or aliasing products, ensuring maximum loudness without side-effects.

Modulation Index offers highly optimized modifications to most current generation FM exciters and STL systems, customized to the individual unit at its operating frequency. These modifications permit the units to pass the most highly processed composite baseband signal while adding less than 1% overshoot — often an improvement of 10:1 or more, resulting in a 1dB loudness advantage with almost any audio processing. This is a very cost effective solution offering better sonic performance than the digital alternative, since there is no data compression to cause distortion. These modifications offer better performance than almost all of the latest analog equipment. Fig. 10 shows the optimized response of the STL System shown in Fig. 3. Fig. 11 shows the optimized response of the FM Exciter shown in Fig. 5.

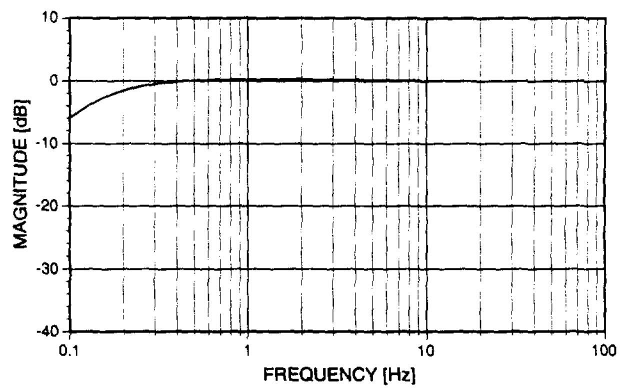

Fig. 10 OPTIMIZED STL SYSTEM 1 RESPONSE
MOSELEY ASSOCIATES, INC. PCL-606/C

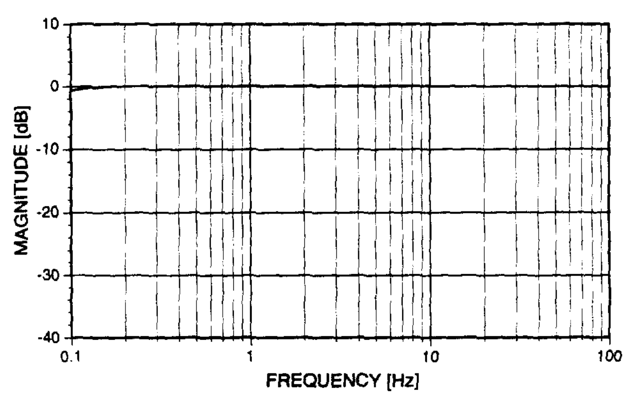

Fig. 11 OPTIMIZED FM EXCITER 1 RESPONSE
CONTINENTAL ELECTRONICS MFG., CO., INC. 802A